# WIDEBAND SPEECH CODING SYSTEM AND METHOD

## CROSS-REFERENCE TO RELATED APPLICATIONS

This application claims priority from provisional applications: Serial Nos. 60/239,731, filed 10/12/00 (TI-31551P), 60/228,215, filed 08/25/00 (TI-31551PS) and 60/206,156, filed 5/22/00 (TI-29772P). ~~The following patent applications disclose related subject matter: Serial Nos. 09/....., filed .... (...). These cross-referenced applications have a common assignee with the present application.~~

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## BACKGROUND OF THE INVENTION

The invention relates to electronic devices, and, more particularly, to speech coding, transmission, storage, and decoding/synthesis methods and systems.

The performance of digital speech systems using low bit rates has become increasingly important with current and foreseeable digital communications. Both dedicated channel and packetized-over-network (VoIP) transmission benefit from compression of speech signals. The widely-used linear prediction (LP) digital speech coding compression method models the vocal tract as a time-varying filter and a time-varying excitation of the filter to mimic human speech. Linear prediction analysis determines LP coefficients a(j), j = 1, 2, ..., M, for an input frame of digital speech samples {s(n)} by setting

$$r(n) = s(n) - \Sigma_{M \geq j \geq 1} a(j)s(n-j) \qquad (1)$$

and minimizing the energy $\Sigma r(n)^2$ of r(n) in the frame. Typically, M, the order of the linear prediction filter, is taken to be about 10-12; the sampling rate to form the samples s(n) is typically taken to be 8 kHz (the same as the public switched telephone network (PSTN) sampling for digital transmission); and the number of samples {s(n)} in a frame is often 80 or 160 (10 or 20 ms frames). Various windowing operations may be applied to the samples of the input speech frame. The name "linear prediction" arises from the interpretation of $r(n) = s(n) - \Sigma_{M \geq j \geq 1} a(j)s(n-j)$ as the error in predicting s(n) by the linear combination of preceding